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SECOND ICTP - URSI - ITU/BDT SCHOOL ON THE USE OF RADIO FOR DIGITAL COMMUNICATIONS IN DEVELOPING COUNTRIES, INCLUDING SPECTRUM MANAGEMENT

(1 - 19 February, 1999)

Performance of Digital Communications Laboratory

Dr. Mike Fitton

Centre for Communications Research University of Bristol Bristol GREAT BRITAIN

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"Performance of Digital Communications Laboratory" M.P.Fitton, University of Bristol, UK

email: mike.fitton@bristol.ac.uk

Overview:

In this laboratory, you will make use of the '*HP I-Q Tutor*' software package in-order to gain a further insight into the performance aspects of modern digital communication systems. The tools allow you to investigate the *bit error rate* performance in terms of received *signal-to-noise ratio*, for a variety of digital modulation methods. Furthermore software demonstrates the impact of channel filtering, multipath channel impairements, and non-ideal RF power amplification. You will investigate constellation plots (I-Q vectors) and eye-diagrams for a variety of channel conditions and modulation schemes.

The Laboratory

It is recommended that you follow the experimental procedure outlined in the HP IQ Tutor *Digital Microwave Communications Guide*. However, if time permits, please consider an evaluation of the other modulation schemes also supported by the software.

To run the software, change directory to 'iqtutor' and issue the DOS command 'iqtutorc'.

It is suggested that delegates don not cover the material on M-curves (pages 5-16 to 5-19), as this is not covered in the lecture material. M-curves useful in analysing microwave links, but do not have widespread use.

<u>Hints</u>

Move about the program using the arrow keys. To enter the EDIT mode, type 'E', and to exit the EDIT mode press <Enter>.

The IQ Vector diagram can be rotated using the 'PgDn' and 'End' keys.

It is recommended that the multipath simulator is switched-off during the evaluation of the non-linear amplifier impairments.

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This section walks you through several experiments and activities designed to give you additional insight into practical digital communications system operation. In some cases you will be able to anticipate the results. In others, the results will be more subtle and obscure. You should find them all interesting.

Refer back to the Benchtop Overview screen. By now you have probably noticed that on the top left of most displays is the message "Ideal P(e) <= xx." The number xx is an approximation of the ideal probability of error based on the signal-to-noise ratio (SNR) and the modulation type. P(e) is a measure of the rate at which received symbols are misinterpreted. Errors can be caused by system noise, clocking errors, or any of a number of problems associated with signal degradation. A probability of error of .01 means that on the average there will be one misinterpreted symbol in every 100 symbols received.

The most common cause of errors in symbol detection is that noise in the system causes a symbol to cross a decision boundary and to be detected improperly. One might conclude from this that for a given modulation scheme and signal-to-noise ratio, an estimate of P(e) might be derived. Signal-to-noise ratio in a digital communications systems is indeed a valuable indicator of system performance. Theoretical curves have been generated for most modulation types that provide estimates of P(e) vs. SNR. The I-Q Tutor program uses these theoretical curves to derive the P(e) number displayed on the screen.

P(e) Versus SNR

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Actual error measurements of a system are time-consuming, especially in modern, low-error systems. It would be advantageous if we could predict errors based on the signai-to-noise ratio, which can be measured much more quickly. This experiment compares the actual errors experienced in the I-Q Tutor system to those predicted by the theoretical curves.

Another more common measure of communications efficiency and quality is Bit Error Rate (BER). BER is a measure of the average number of bits in error divided by the total number of bits received. If the P(e) is known, the BER can be calculated, but it will be a function of design parameters including the number of states in the modulation type, whether or not Grey codes are used, the types of error-correcting schemes employed, etc. Because of this, we will limit our measurements to P(e) in spite of the fact that most-real system measurements will measure BER.

Procedure:

- First power up the system. Use the EDIT mode to set I-Q Tutor to BPSK, Filter Alpha = .1, and SNR = +6 dB. Also make sure that neither the "HPA Impairment On" or "Multipath Fade On" messages are displayed. If these impairments are on, turn them off by using the "D" key to get to the advanced design page and select 15 dB HPA backoff and 0 dB fade depth.
- 2. Once the system has been initialized (the WORKING!!! message disappears), move the cursor to the output of the bit detector and down to the display of the bit detector outputs and their spectrum (the second set of signals from the right). You will examine these signals to determine how many errors are actually being experienced in the system.

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- 3. Then, while looking at the time display of the bit detector output, you will notice errors by looking for dashed lines. Both the detected and transmitted signals are plotted. If an error occurs, the dashed lines indicating the transmitted signal will be visible. The number of errors is tabulated in the display label. Sometimes when using more complex modulation types, the displayed number of errors will be higher than the number of errors that you can see. This is because the error may also occur in the Q channel and you only see the I channel displayed. Since BPSK is transmitted entirely on one channel, you should be able to observe all of the errors. To calculate the P(e), you will divide the number of displayed errors by the number of transmitted symbols which is 100.
- 4. Write the P(e) you have calculated in the table below and plot the P(e) on the P(e) vs. SNR graph provided. Now re-edit the parameters reducing the SNR by 2 dB and repeat the procedure. Do this until you have plotted the P(e)s for SNR's down to -2 dB.

BPSK SNR Errors Errors/100 = P(e)6dB 4dB 2dB 0dB -2dB dB 10 P(e) 5 0 30 - 10 ° ъ 20 25 10 ⁰ 10 ** 10 -1 BPSK 10-2 10 -2 10⁻³ 10 ⁻³ . 10~1 10 🗝 10-5 10 -5 -5 0 5 ۳Ô ъ 25 20 30 đΒ SNR

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Figure 5-1. P(e) vs. SNR for BPSK

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- 5. There is probably some discrepancy between your findings and the theoretical curves. This is due to two things. First, you are measuring a random process which by its nature is unpredictable. Second, you are using very small samples to determine your results. At any rate, the results you obtain should resemble the theoretical results enough to persuade you to at least believe in the overall picture they paint. The most important lesson to learn from all of this is that small increases in SNR can be catastrophic to the quality of transmission in digital communications systems.
- 6. If you wish to verify P(e) vs SNR for other modulation types, the theoretical curves are given below. The procedure is basically the same. You will need to use a different range of SNRs when collecting your data so that there are a reasonable number of errors to observe. Use the values shown in the tables of Figure 5-2a.

QPSK	SNR	Errors	Errors/100 = P(e)
·	8dB		
	6dB		
	4dB		
	2dB		
	OdB		

Figure 5-2a. QPSK Table of Errors

160AM

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SNR	Errors	Errors/100 = P(e)
16dB		
14dB		
12dB		
10dB		
8dB		





Figure 5-2c. Theoretical P(e) Curves

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P(e) Versus Filter Bandwidth You may recall that there are two reasons for filtering the data in a digital communications system. The first is to limit bandwidth. It should be clear from looking at the baseband I and Q signals that we indeed limit bandwidth quite a bit. The second reason is to reduce the noise introduced in transmission. This experiment is directed at determining whether or not the filter makes any difference in the actual system's tolerance to noise. The measure we will use to determine overall transmission quality is P(e), the rate at which misinterpreted symbols or states are received. A P(e) of .025 means that in 1000 symbols received, on the average 25 would be in error.

Procedure:

- Use the EDIT function to set up the I-Q Tutor system to BPSK, Filter Alpha =.9, and SNR = 2.
- Move the cursor to the output of the bit detector and down to examine the time display of the output. From here it is easy to see actual errors in the detected signal since they appear as deviations from the dashed line representing the transmitted data.
- 3. Using the displayed error count (since you can't see the errors in the Q channel), divide by 100 to obtain P(e) for the system setup given above. Record your results in the table below. It's important that you calculate the P(e) from the error count and not just read P(e) in the upper left part of the screen. The "Ideal P(e)" represents an average value and won't show you the P(e)'s dependence on the filter alpha.
- 4. Now change the filter alpha to 0.1 and find the new resulting P(e). Switch the filter alpha back to 0.9 and take another P(e) measurement. Continue doing this until you have taken five separate measurements for both filter alphas. The reason for switching the filter alpha back and forth is to obtain a large enough statistical sample to obtain meaningful results.
- 5. After you have filled the table, take the average of each set of data and write the average value of P(e) for each filter alpha in the bottom of the table. There should be a difference between the two numbers indicating that the narrower alpha = .1 filter

_ reduces the number of errors seen at the receiver.

What is happening here? As the filter alpha is reduced, the total system bandwidth is also reduced, but the amount of information is kept constant. The only real change is that with higher alphas, more redundant information is transmitted and more noise is allowed to enter the system. This results in slightly higher P(e)s for the same SNR's.

FILTER	Alpha = .1	Alpha = .9
Errors		
AVERAGE		

Figure 5-3. Table of Measured Symbol Error Values

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EN PARTS

Noise, Errors and the I-Q Vector Diagram

Perhaps you noticed that there seems to be a difference in the number of errors that different modulation types experience with the same SNRs. For any given SNR, the modulation scheme will directly influence the probability of error. Why can't we send 16QAM just as easily as BPSK? In other words, why do we have to use more power to obtain the same P(e) with 16QAM than BPSK? The reason becomes clear when we examine the two signals in the I-Q Vector display using the constellation plot.

Procedure:

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- 1. Set up the I-Q Tutor system using the EDIT function to BPSK, SNR=40 dB and Filter Alpha = .5. Be sure the path impairments are turned off.
- Move the cursor to the output of the I-Q demodulator and down so that you can see the time domain display of the baseband I-Q signals.
- 3. Now enter the I-Q Vector display by pressing the "ENTER" key.
- To look at the data at the clock sampling instant press the "/" key.
- You should see a practically noise-free diagram of a BPSK signal states. Notice that the states are very small points and that they are very clearly separated from each other.
- 6. Return to the time domain display and spectrum by pressing the "ENTER" key again.
- 7. Introduce some noise to the system. Press the "EDIT" key again to re-enter the edit mode and change the SNR to 8 dB.
- 8. When the WORKING!!! message disappears, re-enter the I-Q Vector display, select the constellation display, and notice that though the states are still clearly separate, there is some visible noise on the signals. The noise increases the ambiguity of the signal by spreading out the states.

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- 9. Return to the time domain display and re-EDIT the parameters, changing the modulation type to 16QAM while leaving the rest of the parameters unchanged.
- 10. Press the "ENTER" key to see the effect of the same amount of noise on a 16QAM signal. As you can see, the visible noise has not increased. However, the states were closer together to start with so the same noise causes greater ambiguity in the signal. This translates into a higher P(e).

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Multipath and "M" Curves

In this lab you will explore what multipath distortion is all about in more depth. With a little patience, the experiments in this section will give you an intuitive feel for what multipath distortion looks like in the frequency, time, and vector domains. In addition, you will learn how to characterize multipath with what are known as "M" curves.

Multipath distortion is probably one of the most troublesome problems for both designers and users of digital radios. As modulation formats get more complex, even slight amounts of multipath will crash a system without sophisticated compensation techniques.

A simple model for multipath distortion assumes that the signal travels over two different paths to the receiver - a direct path, and one that is delayed by a small amount due to reflection or refraction. This delay causes notches in the frequency response of the received signal. If the delay is T microseconds, the notches occur every 1/T MHz. The depth of the notch is determined by the relative amplitude of the main and delayed signals. For example, if the amplitude of the delayed signal is 90% of the amplitude of the main signal, the notch will be approximately 20 dB deep. If the amplitude of the delayed signal is 99% of the main signal, the notch will be 40 dB deep. If the two signals are of equal magnitude, the notch will be infinitely deep.

Let's investigate this characteristic - first in the frequency domain.

Procedure:

- 1. Position the probe pointer to look at the output of the I/Q demodulator, and then look at the time display of the output. Use the Edit function to set up I Q Tutor for QPSK, SNR=40dB and Filter Alpha=.3. DO NOT HIT THE ENTER KEY YET!
- 2. Now enter the advanced design screen by pressing the "D" key. Press the up arrow to change the delay to 16.6ns. You are now changing the delay between the main and the delayed signals. Note that this corresponds to creating notches every 1/16.6ns Hz, or every 60 MHz.
- 3. Press the right arrow twice to edit the notch depth. Press the up



arrow key repeatedly to change the depth to 40 dB. Note that you are changing the relative magnitudes of the main and delayed signals. Observe the notch as you change the depth.

- Press the left arrow once again to edit the notch position. Press the up arrow key repeatedly to change the position to +100%. You are changing the phase of the delayed signal to move the notch.
- 5. Draw the frequency response curve of the delay in the box below. The I-Q Tutor is set up to simulate digital transmission at a symbol rate (baud rate) of 30 MHz. Therefore, Fs and -Fs are at +30MHz and -30MHz respectively, and the notches actually fall at 60 MHz intervals.





Changing the Delay Time

6. Now, press one left arrow to modify the multipath delay again. Hold the up arrow key down until you reach 35ns delay. Draw the frequency response curve below. Note that your notches are now occurring every 1/35ns or 29 MHz.



7. Change the delay back to 6.3ns. This is a very commonly used delay for multipath simulation and is derived from empirical field testing of actual multipath fades. Draw the frequency response curve below.





EXPERIMENTAL EXERCISES

Multipath Distortion in the Time and Vector Domains

- 8. Change the notch depth to 10 dB. Now make sure that you have the following multipath settings:
 - Delay: 6.3 ns Depth: 10 dB Position: +100%

Press the ENTER key to exit the advanced design mode and get back to the edit mode. Make sure that the edit parameters are as follows:

Modulation Type: QPSK SNR: 40dB Filter Alpha: .3

Now press the ENTER key again to exit the edit mode and recompute the system waveforms. When the "WORKING!" message goes away, you should be looking at the time waveform of the signal after the I/Q demodulator.

Press the ENTER key to look at the Vector Diagram. Note that the vector diagram is rotated due to the multipath distortion. Using the knob on the computer, rotate the diagram about 45 degrees clockwise, until the sides of the QPSK pattern are as parallel as possible to the sides of the computer screen. Now, use the left arrow (18 times) to rotate the diagram to the left until you see an eye diagram. Draw the eye diagram below. Notice that the multipath distortion introduces some ISI (inter symbol interference) in the eye diagram. Real radios automatically correct for the phase rotation, but the ISI can't be fixed without equalization. This is what causes system degradation and poor BER performance.



"M" Curves

The effect of multipath distortion on system performance is usually characterized in what is called an "m" curve. A fixed multipath delay is chosen (say 6.3 ns) and the BER is measured as a function of notch depth and notch position. To make the measurements more rapidly, a fixed BER is chosen at a relatively high level (say 10^{-3}) and the notch depth required to reach that fixed BER is plotted as a function of notch position. A theoretical example of such a measurement appears in Figure 5-4. Note that over the notch position span of -Fs to +Fs, the notch depth required to reach a 10^{-3} BER level follows an "m" shaped curve hence the name of this measurement.





Figure 5-4. M Curve for QPSK for 6.3 nsec delay

Let's try to crudely recreate an "m" curve. Although BER measurements do not work in I-Q Tutor when multipath distortion is used, we can look at the shape of the eye diagram to guess at the signal degradation. Our job will be to change system parameters so that the signal degradation will always took approximately the same (approximating constant BER).

9. Change the notch position to 0%. (Remember, to get back to the advanced design screen you need to press the Enter key to get back to the time display; the Edit or "E" key to get the edit mode; and the "D" key to get the advanced design screen). After you have changed the notch position, press the Enter key twice - once to display mode and calculate new waveforms. After the "WORKING!" message disappears, press Enter to see the eye diagram and press 18 right arrows to rolate back to a vector



diagram. Notice that a 10dB deep notch centered in the data spectrum significantly reduces the signal amplitude. Press 15 ">" (greater than) keys to increase the signal amplitude. Note also that when the notch is centered, the pattern is no longer rotated with respect to the I and Q axes. Use the knob to rotate the pattern back to 0 degrees orientation. Press the left arrow key 18 times until you see the eye diagram again. Press the "S" key to store this eye diagram. Note that the eye diagram is significantly more degraded (higher BER) when the notch is centered in the data spectrum. We will call this picture our "Reference BER". Plot a point on the graph below at zero offset and notch depth of 10dB. This means that it takes a 10 dB notch depth at zero offset to create our "Reference BER" picture.



10. Now go back to the advanced design screen (refer to the beginning of step 9) and change the notch position to 30% and the notch depth to 7dB. Exit the advanced design screen and

edit mode. Press enter to get an eye diagram and press the "<" (less than) key 4 times to adjust the system gain. Rotate your eye diagram back to a vector diagram. Use the knob to rotate the vector diagram until the sides of the QPSK pattern are as parallel as possible to the sides of the computer's CRT screen (approximately 25 degrees clockwise). Now rotate back to an eye diagram. Note that this eye diagram is roughly the same as our "Reference BER" picture. To prove this, recall the "Reference BER" picture by pressing the "R" key. You can see the new eye pattern again by pressing any other key. Note that the system is most sensitive to multipath distortion when the notch is about 30% of Fs - it requires a smaller notch depth of 7dB to get the same amount of degradation. "Plot a point on your graph at 30% offset and 7dB notch depth.

11. Go back to the advanced design screen one final time and change the notch position to 100% and the notch depth to 30 dB. Exit the advanced design screen and edit mode. Press Enter to get an eye diagram and press the "<" (less than) key 5 times to adjust the system gain. Rotate your eye diagram back to a vector diagram so that you can align the QPSK pattern so its sides are parallel to the computer screen (approximately 50 degrees clockwise). Now rotate back to an eye diagram. Compare it to the "Reference BER" picture by pressing the "R" key. Does it look roughly the same? When the notch is positioned outside the data spectrum, the system is much more tolerant of multipath distortion. It requires a much greater notch depth to create the same amount of degradation. Finish this exercise by plotting a point on your graph at 100% offset and 30dB notch depth. Congratulations! - you have just plotted a rough "m" curve!

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Distortion Caused by High Power Amplifier Nonlinearities

Output power is a precious commodity in digital microwave systems. In order to get best Signal to Noise ratio, system designers often run their High Power Amplifiers (HPA's) near their maximum rated output (the compression range). In this lab, you will investigate the tradeoffs that occur when HPA's are operated in compression. Specifically, you will see what an HPA's characteristics are in the compression range and you will see what effect these characteristics have on digital microwave signals.

Procedure:

- 1. Move the probe pointer to the output of the demodulator and then press the down arrow to look at the time domain display of the demodulated I and Q signals.
- 2. Enter the Edit mode by pressing the Edit or "E" key and change the modulation type to 16QAM. (Make sure that the SNR=40 and filter alpha=.3). Press the Enter key to exit the Edit mode and recalculate the waveforms.
- 3. Once the "Working!!!" message disappears press the enter key again to look at the Vector Display of the 16QAM modulation. Press 5 ">" (greater than) arrows to adjust the gain, then press the "/" key to look at a Constellation diagram. You are now looking at the constellation diagram of an undistorted 16QAM signal.
- 4. Store this constellation diagram by pressing the "S" key. We will be comparing the distorted constellations to this diagram later.
- Press the Enter key to get back to the waveform time display. Then press the Edit or "E" key to get to the edit mode. Press the "D" key to enter the Advanced design screen to change the HPA parameters.
- 6. The large box on the right hand side of the screen describes the HPA characteristics. In the large box you see two displays. The upper one shows the amplitude at the output of the HPA vs. the amplitude at the input to the HPA. Ideally this should be a straight line with slope equal to the gain of the HPA. In the

bottom box you see the phase of the signal at the output of the HPA plotted against the input level. We arbitrarily call the phase of the output signal, measured at low levels, 0 degrees phase. Ideally, this phase should not change with input amplitude.

- 7. At the top of the large box, you will see the amount of HPA backoff the amount the input signal is reduced or "backed off" from the HPA's maximum output. This number should be -15dB, the maximum backoff in this exercise. Note that at this backoff, we are operating in a very linear region of the amplifier, and the graphs of output phase vs. input amplitude (AM to PM) are ideal.
- 8. Press a left arrow to edit the HPA backoff and then press up arrows to change the backoff to -10dB. Notice what happens to the output phase vs. input amplitude graph. The amplifier is gradually moving into its nonlinear region which causes some unwanted (or incidental) phase deviation.
- Now press up arrows to change the backoff to -3dB. Notice the actual amplitude compression in the upper graph of output amplitude vs. input amplitude. Draw the curve in the box provided below.





10. Look at the lower graph of phase shift vs. input amplitude. Notice that there are almost 30 degrees of phase shift at maximum input level. Draw the curve in the box provided below.

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- 11. Now, let's observe the effect that a 3dB backoff has in the Vector Domain. Press the Enter key once to exit the advanced design screen and return to Edit mode. Press the Enter key again to exit the Edit mode and recalculate the waveforms.
- 12. Wait until the "Working!!!" message disappears (about 50 seconds). You should now be looking at the time display at the demodulator output. Press the Enter key to look at the Vector Display. Press 2 ">" (greater than) arrows to adjust the gain and press "/" to see a Constellation diagram. Compare this diagram to the undistorted diagram by pressing the "R" key to recall the undistorted diagram.
- 13. You should observe several distortions. First, you will see that the outermost 4 states have collapsed slightly to the center of the vector diagram. Why those states? Look at the first graph you drew above of output amplitude vs. input amplitude. Notice that at higher input amplitudes, the output amplitude no longer increases linearly with the input. Instead, the output amplitude stops increasing as the amplifier goes into compression. This is

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what causes the outermost states to look squashed compared to the inner ones.

- 14. Second, look at the phase rotation of the inner 4 states vs. the outer 12 states. Note that the inner states are rotated counter-clockwise with respect to the outer states. Why does this happen? Again, refer to the second graph you drew above showing output phase vs. input amplitude. Note that the states with larger amplitudes are rotated more than the smaller amplitude states. Some of you may ask why the outer states are not rotated by 30 degrees as the graph shows. The reason is that the program automatically normalizes, or rotates, the diagram so that the outermost 4 states are always parallel to the sides of the display screen. This simulates the corrective effects of some types of carrier recovery schemes.
- 15. Finally, press the "/" key to return to Vector Diagram mode and press 18 left arrows to look at the eye diagram. Notice that the HPA nonlinearity also adds to the Inter-Symbol Interference as can be seen by the partial eye closure of the diagram.

These distortions also show up in the frequency domain. An unwanted side-effect of HPA distortion is that the frequency spectrum is broadened with unwanted distortion sidebands - possibly spilling over into adjacent channels, causing interference there as well.

16. We will now look at these distortion sidebands. Press the Enter key to return to the time domain waveform of the IQ demodulator output. Press one up arrow to get to the system overview picture. Now press two left arrows to move the probe pointer to the microwave portion of the system. Press a down arrow to look at the frequency domain waveforms of the transmitted and received spectrums. Compare the transmitted spectrum (obtained from signals just before the transmitter's High Power Amplifier) to the received spectrum below. Note that the spectra of the received signals are all a little wider at their bases due to the distortion in the waveform. You can try even less backoff (such as 0 dB) to see an even larger effect on the frequency spectrum.

You have now seen that although getting as much output power from your system as possible is important, it must be traded off with waveform distortions due to HPA nonlinearities as the amplifier operates in its compression range. The distortions due to HPA nonlinearities include compression of the outer states of the signal, rotation of inner states with respect to the outer states, and ISI - all of which combine to cause the system to be more error prone.

Conclusions:

As we have seen, there are large differences among the various modulation types in terms of their tolerances to noise. In general, the more complex the modulation type or the greater number of states, the less tolerant that signal will be to noise. This has been formally stated in what is known as the Shannon Limit Theorem. Having seen the noise on different signals, it should be fairly straight-forward to understand why this is true. As modulation complexity increases, the actual signal level difference between states decreases. As this difference decreases, it becomes more and more likely that the noise level will exceed this difference. This ability to choose tolerance to noise at the expense of information carrying capacity makes digital communications attractive for many long hauf applications.